Amendments to the Claims

The listing of claims will replace all prior versions, and listings of claims in the application:

1. (*previously presented*) A method for reducing overhead, latency, and/or packet loss for a voice and data over Internet Protocol (VoIP) packet transmitted between an originating gateway and a destination gateway, comprising the steps of:

(1) compressing voice streams and/or data streams from a plurality of concurrent calls from a plurality of channels into packets;

(2) aggregating said packets into a single packet to produce the VoIP packet, said VoIP packet including information, that when executed, synchronizes a current channel state at the originating gateway with a record of said channel state at the destination gateway; and

(3) transmitting the VoIP packet between the originating gateway and the destination gateway through a single virtual connection.

2. (*previously presented*) The method of claim 1, wherein step (2) further comprises the step of providing a plurality of voice frames and/or data frames and a plurality of header frames in the VoIP packet, wherein said plurality of header frames comprises at least one header frame selected from the group consisting of a time stamp header, local network address header, IP address header and UDP header and at least one

header frame selected from the group consisting of a version number header and control information header.

3. *(previously presented)* The method of claim 1, further comprising the step of converting analog voice streams and/or analog data streams to digital voice streams and/or digital data streams prior to executing said compressing step.

4. *(previously presented)* The method of claim 1, further comprising the step of transmitting a check sequence packet at regular packet intervals, wherein a duration of said intervals is altered to reach a desired tradeoff between increased tolerance to loss and bandwidth, wherein a parity system and information located inside of said check sequence packet is used to regenerate missing or damaged information in a previously transmitted VoIP packet.

5. *cancelled*

6. *(previously presented)* A system for reducing overhead, latency, and/or packet loss in a voice and data over Internet Protocol (VoIP) packet transmitted between an originating gateway and a destination gateway, said system comprising:

media framing means for aggregating packets from a plurality of concurrent calls from a plurality of channels into a single packet to produce a VoIP packet;

transmission control means for providing information in the VoIP packet to synchronize a current channel state at the originating gateway with a record of said channel state at the destination gateway;

redundancy means for regenerating missing or damaged information transmitted in the VoIP packet; and

a single virtual connecting means for transmitting the VoIP packet from the originating gateway to the destination gateway.

7. (*previously presented*) The system of claim 6, wherein the VoIP packet comprises a plurality of voice frames and/or data frames and a plurality of header frames, comprising at least one header frame selected from the group consisting of a time stamp header, local network address header, IP address header and UDP header and at least one header frame selected from the group consisting of a version number header and control information header.

8. (*previously presented*) The system of claim 6, further comprising:

means for transmitting and receiving voice streams and/or data streams;

means for converting analog voice streams and/or analog data streams to digital voice streams and/or digital data streams;

means for compressing digital voice streams and/or digital data streams into said packets; and

means for transmitting a check sequence packet after a transmission of a predetermined quantity of VoIP packets, wherein said media framing means produces each VoIP packet of said predetermined quantity of VoIP packets.

9. (*previously presented*) The system of claim 8, wherein said check sequence packet includes check sequence information that, when executed, regenerates missing or damaged information transmitted in the VoIP packet, wherein said redundancy means produces said check sequence information.

10. *cancelled*

11. *cancelled*

12. (*previously presented*) A computer program product comprising a computer useable medium having computer readable program code means embedded in said medium for causing a computer to reduce overhead, latency, and/or packet loss in a VoIP packet transmitted between an originating gateway and a destination gateway, comprising:

first computer readable program code means for causing the computer to compress voice streams and/or data streams from a plurality of concurrent calls from a plurality of channels into packets;

second computer readable program code means for causing the computer to aggregate said packets into a single packet to produce the VoIP packet;

third computer readable program code means for causing the computer to transmit the VoIP packet between the originating gateway and the destination gateway through a single virtual connection;

fourth computer readable program code means for causing the computer to provide information in the VoIP packet to synchronize a current channel state at the originating gateway with a record of said channel state at the destination gateway; and

fifth computer readable program code means for causing the computer to determine if the VoIP packet contains missing or damaged information or to regenerate said missing or damaged information.

13. *(previously presented)* The computer program product of claim 12, wherein said second computer readable program code means further comprises computer readable program code means for causing the computer to provide a plurality of voice frames and/or data frames and a plurality of header frames in the VoIP packet, wherein said plurality of header frames comprises at least one header frame selected from the group consisting of a time stamp header, local network address header, IP address header and UDP header and at least one header frame selected from the group consisting of a version number header and control information header.

14. *(previously presented)* The computer program product of claim 12, further comprising sixth computer readable program code means for causing the computer to convert analog voice streams and/or analog data streams to digital voice

streams and/or digital data streams prior to executing said first computer readable program code means.

15. (*previously presented*) The computer program product of claim 12, wherein said fifth computer readable program code means further comprises computer readable program code means for causing the computer to transmit a check sequence packet upon completion of an execution of said third computer readable program code means, wherein said check sequence packet comprises information that, when executed, regenerates said missing or damaged information.

16. *cancelled*

17. *cancelled*

18. (*previously presented*) The method of claim 1, wherein said channel state identifies whether a channel is open or on-line.

19. (*previously presented*) The method of claim 1, wherein step (2) further comprises the step of providing in the VoIP packet a channel present header for indicating whether a channel is currently open and communicating.

20. (*previously presented*) The method of claim 1, wherein step (2) further comprises the step of providing information in the VoIP packet to instruct the destination gateway to start using said record to deframe the VoIP packet.

21. (*previously presented*) The system of claim 6, wherein said single virtual connecting means enables transmission of the VoIP packet from said media framing means at the originating gateway directly to a second media framing means at the destination gateway.

22. (*previously presented*) The system of claim 6, wherein said single virtual connecting means enables transmission of the VoIP packet from said transmission control means at the originating gateway directly to a second transmission control means at the destination gateway.

23. (*new*) The method of claim 1, wherein step (3) further comprises the steps of:

(a) transmitting a plurality of VoIP packets from the originating gateway to the destination gateway; and

(b) transmitting a check sequence packet from the originating gateway to the destination gateway upon completion of a transmission of said plurality of VoIP packets, wherein said check sequence packet comprises information that, when executed, regenerates missing or damaged information transmitted in any of said plurality of VoIP packets.

24. (*new*) The system of claim 6, wherein said redundancy means further comprises:

means for transmitting a check sequence packet upon completion of a transmission of a predetermined quantity of VoIP packets; and

means for regenerating missing or damaged information in any of said predetermined quantity of VoIP packets.

25. (*new*) The system of claim 24, wherein said redundancy means further comprises means for implementing a parity system to regenerate missing or damaged information.

26. (*new*) The computer program product of claim 12, wherein said fifth computer readable program code means further comprises:

sixth computer readable program code means for causing the computer to transmit a check sequence packet at regular packet intervals;

seventh computer readable program code means for causing the computer to alter a duration of said intervals to reach a desired tradeoff between increased tolerance to loss and bandwidth; and

eighth computer readable program code means for causing the computer to regenerate missing or damaged information in a previously transmitted VoIP packet by using information located inside of said check sequence packet.

27. (*new*) The computer program product of claim 26, further comprising ninth computer readable program code means for causing the computer to utilize a parity system to regenerate missing or damaged information.